

CLAIMS

WHAT IS CLAIMED IS:

1. An audio conferencing system comprising:
at least one loudspeaker for converting a first electrical signal into sound;
a plurality of conference stations in spaced relation, each conference station comprising a directional microphone for converting sound into a directional microphone signal, the directional microphone signals collectively forming a plurality of directional microphone signals; and
a signal processor for modifying at least one of the plurality of directional microphone signals and a receive signal, the signal processor producing at least one of a transmit signal and the first electrical signal.
2. The audio conferencing system of claim 1 wherein the modifying comprises an algorithm to perform acoustic echo cancellation.
3. The audio conferencing system of claim 1 wherein the modifying comprises an adaptive beamforming technique.
4. The audio conferencing system of claim 3 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.
5. The audio conferencing system of claim 1 wherein the modifying combines the plurality of directional microphone signals in order to selectively attenuate or amplify a sound source.

6. The audio conferencing system of claim 1 wherein the modifying selects for separate processing at least two groups of directional microphone signals from the plurality of directional microphone signals.

7. The audio conferencing system of claim 6 wherein the modifying of each of the at least two groups uses an adaptive beamforming technique.

8. The audio conferencing system of claim 1 further comprising at least one omni-directional microphone for converting a sound field into an omni-directional microphone signal.

9. The audio conferencing system of claim 8 wherein the modifying comprises combining at least one of the plurality of directional microphone signals and the at least one omni-directional microphone signal, based upon at least one room condition.

10. The audio conferencing system of claim 9 wherein the at least one room condition comprises at least one of background noise, a level of acoustic echo, and the detection of side conversations.

11. The audio conferencing system of claim 1 wherein each of the conference stations comprises a transducer for producing an acoustic test signal.

12. The audio conferencing system of claim 1 wherein the signal processor uses a test signal to determine at least one of microphone and room acoustic characteristics.

13. The audio conferencing system of claim 1 wherein the contribution to the transmit signal of a selected sound source relative to other sound sources may be increased or decreased from a location remote from the audio conferencing system.

14. The audio conferencing system of claim 1 further comprising an interface compatible with a communication network, the interface coupling the transmit signal to the communication network, and the communication network to the receive signal.

15. The audio conferencing system of claim 14 wherein the communication network is a packet network.

16. The audio conferencing system of claim 1 further comprising a manual input device used for at least one of controlling calls and entering system parameters.

17. The audio conferencing system of claim 1 wherein the signal processor is a digital signal processor.

18. A method of operating an audio conferencing system comprising:
receiving a first electrical signal; /
transducing each of a plurality of sound fields into a microphone signal, the microphone signals collectively forming a plurality of microphone signals;
processing at least one of the plurality of microphone signals and the first electrical signal to produce a second electrical signal; and
transmitting the second electrical signal.

19. The method of claim 18 wherein the processing comprises an algorithm to perform acoustic echo cancellation.

20. The method of claim 18 wherein the processing comprises an adaptive beamforming technique.

21. The method of claim 20 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.

22. The method of claim 18 wherein the processing comprises selecting at least two groups of microphone signals from the plurality of microphone signals.

23. The method of claim 22 wherein each of the at least two groups of microphone signals is used in a separate adaptive beamforming arrangement.

24. The method of claim 18 wherein the processing uses at least one parameter representative of at least one of a microphone acoustic characteristic, a transmission delay, and an acoustic characteristic of a room.

25. The method of claim 18 wherein the processing may be modified remotely during operation.

26. The method of claim 18 wherein the processing is performed using a digital signal processor.

27. The method of claim 18 wherein at least one of the first electrical signal and the second electrical signal are a digital signal.

28. The method of claim 27 wherein at least one of the first electrical signal and the second electrical signal are compliant with a packet protocol.

29. The method of claim 18 further comprising:
generating a first electrical test signal;
converting the first electrical test signal to an acoustic test signal at a first location;
sampling the acoustic test signal at a second location;
transforming the sampled acoustic test signal into a second electrical test signal; and
deriving at least one of a microphone acoustic characteristic, a transmission delay, and an acoustic characteristic of a room using the second electrical test signal.

30. A method of operating an audio conferencing system comprising:
receiving a plurality of microphone signals; /
selecting at least two groups of microphone signals from the plurality of microphone signals;

processing each of the at least two groups of microphone signals using an adaptive beamforming technique, the processing producing an output signal for each of the at least two groups of microphone signals; and
combining the output signals.

31. The method of claim 30 further comprising:
performing acoustic echo cancellation on at least a portion of the plurality of microphone signals.

32. The method of claim 30 wherein the selecting is based upon at least one of an amplitude of a microphone signal, a propagation delay, and an input from a user.

33. The method of claim 30 wherein the adaptive beamforming technique comprises at least one of a normalized least mean squares algorithm and a recursive least squares algorithm.